## REMARKS

Claims 1-12 are pending in the application. Claims 1-12 are rejected.

In the drawings figures 1 and 14C have been objected to. Attached is proposed correction to Figs. 1 and 14C as shown in red on the marked-up copy of the drawings. Specifically, Fig. 1 has been designated with the legend "Prior Art" and Fig. 14 is corrected as proposed by the Examiner. The applicant proposes to submit formal substitutes upon notice of allowance.

The Abstract has been objected to. A substitute abstract is attached herewith. The disclosure has been objected to for informalities. The disclosure has been amended as suggested by the Examiner. No new matter has been added.

Claims 1, 5 and 9 are rejected under 35 U.S.C. § 102(e) as being anticipated by Morii (U.S. patent 6,205,421). Claims 1, 5 and 9 have been cancelled herein obviating the rejection.

Claims 2-4, 6-8 and 10-12 are rejected for obviousness over Morii in view of U.S. patent 4,701,955 to Taguchi.

Claims 2, 6 and 10 have been amended to independent form.

The Morii speech coding apparatus includes a speech-analyzing unit 21, which processes digital signals from the input speech receiving unit. A set of characteristic parameters for each of the digital analyzing signals are calculated and that set indicates that the digital analyzing signal may correspond, for example, to a vowel stationary portion of a vowel of the voice or a vowel transitional portion of a vowel of the voice (see column 18, lines 17-26). The Office Action has considered the coding unit of Morii as a rate setting unit similar to and functioning as the rate setting unit as claimed. The coding module 143 codes the bit rate 4Kbps for a stationary vowel.

Taguchi teaches a speech coder which uses LSP (Line Spectrum pair) as speech spectrum information. The Taguchi speech coder has a memory storing LSP information for each of the given length frames for speech data prepared beforehand as a reference pattern and a pattern matching unit which calculates a <u>distance</u> between LSP information in a representative vector and LSP information of the reference pattern to output a label signal indicating the reference pattern having minimum distance.

Morii proposes to teach reducing the rate, in association with a variable rate speech encoding, during a period in which a subject speech is of a voiced sound and it is in a steady state.

Morii however is different from the present invention with respect to the judging method. Morii employs a neural network to constitute its judging method. In particular, the method of Morii is associated with a configuration in which 17 input parameters are to be inputted, one layer is provided as the intermediate layer and five kinds of outputs are produced at the outputting layer. One of these outputs corresponds to the mode associated with the steady state of a voiced sound.

Morii describes about 17 different kinds of input parameters including ones with indications such as LPC and power information. However Morii does not teach nor is concerned with a parameter associated with LSP.

Taguchi is concerned with an objective of reducing the required bit rate by employing a variable transmission frame. It is associated with a method according to which, distinctions between voiced and voiceless portions and between flat and inclined portions of an inputted speech are determined when selecting the transmission frame, and no frame is transmitted during the period corresponding to the voiceless portions or the flat portions of the speech.

(i) According to the analysis method associated with Taguchi, for the DP matching mode, 20 frames (200 ms) are analyzed at a time.

(ii) It involves a step of performing a pattern matching with a reference pattern and calculates LSP distance only from equations (2) and (14).

In contrast, according to the present invention, for an inputted speech, the LPC alignment is investigated to determine whether the concerned portion of a speech is of a vowel or a consonant based on the presence/absence of an LPC aligning more closely than a preassigned distance value and the bit rate for vowel portions is reduced.

Applicant's claim 2, of the present application is concerned with changing the bit rate or a speech portion relating to the distinction between the vowel and consonant portions of a voice.

Claim 2 additionally recites: an LSP coefficient calculating unit calculating an LSP coefficient obtained from the voice signal; and

an LSP interval judging unit judging whether an interval between the LSP coefficients is equal to or less than a prescribed threshold value.

Similarly the independent claims of the present application are concerned with determining the distinction between vowel and consonant portions <u>based on the distance between the associated LPCs</u>.

Applicant's claimed invention of distinguishing every portion of a voice between vowel and consonant portions is different from the pattern matching disclosed in Taguchi. Applicant respectfully points out the fact that it would not be apparent that a certain pair of LPCs among LPCs associated with a speech portion are located very closely when performing the pattern matching based on the distances calculated by employing equations (2) and (14) as disclosed in Taguchi.

It is respectfully submitted that none of the cited documents disclose the technology according to which the distinction between vowel and consonant portions is determined based on the closeness between a pair of LPCs and depending on this distinction, the bit rate is changed.

The distinction with respect to the analysis method between the present invention and the combination of Morii and Taguchi is further expressed as follows:

A method, which adopts a DP matching process, as explained in (i) above, is associated with a problem related to real time communication quality. This quality issue is attributable to a large delay that is inherent with a process in which a certain number of frames are accumulated and processed in a batch-wise manner.

In contrast, according to the present invention, the transmission rate reduction is accomplished without involving such a real time communication quality problem because the method requires only determining the variance with respect to the LSP between the current and the immediately preceding frames. In addition to this advantage, it is possible to achieve more accurate determination according to the method of the present invention because the present invention method relies only on the closeness of LSP parameters associated with the immediately preceding frame in comparison to a method employing the analysis method, as explained as (ii) above, calculating the LSP distances only with equations (2) and (14).

Thus, claims 2, 6 and 10, as amended, are associated with features that are not taught or anticipated by any of the cited references with respect to the method for identifying the distinction between vowel and consonant portions and are patentably unique.

Attached hereto is a marked-up version of the changes made to the specification and claims by the current amendment. The attached page is captioned "Versions with markings to show changes made."

In view of the amendments and remarks set forth above, this application is in condition for allowance which action is respectfully requested. However, if for any reason the Examiner should consider this application not to be in condition for allowance, the Examiner is respectfully requested to telephone the undersigned attorney at the number listed below prior to issuing a further Action.

Any fee due with this paper may be charged to Deposit Account No. 50-1290.

Respectfully submitted,

Brian S. Myers
Reg. No. 46,947

**CUSTOMER NUMBER 026304**

Katten Muchin Zavis Rosenman
575 Madison Avenue
New York, NY 10022-2585
(212) 940-8703
Docket No.: FUJO 19.398 (100794-00170)
BSM:fd

## VERSION WITH MARKINGS TO SHOW CHANGES MADE

IN THE SPECIFICATION:

**At page 3, last paragraph, last line, please change to read as follows:**

The residual signal calculating section 16 generates a residual signal from the input signal by eliminating the vocal [track] tract component determined by the LPC. This residual signal is inputted to the adaptive codebook searching section 17. The adaptive codebook searching section 17 vector-quantizes using an adaptive codebook and quantizes the pitch component of the residual signal. When searching for this adaptive codebook, the adaptive codebook searching section 17 obtains an LPC before quantization and an LPC after quantization from the LPC-LSP converting section 13 and LSP-LPC converting section 15, respectively, in order to select an optimal vector for minimizing the error and performs an error minimization operation. Then, the adaptive codebook searching section 17 transmits he vector-quantized pitch component as a transmitting signal. The remaining signal component obtained by eliminating the pitch component from the residual signal is inputted to the fixed codebook searching section 18. The fixed codebook searching section 18 vector-quantizes the remaining signal obtained by eliminating both [track] tract and pitch components from the input signal and transmits the signal as an output signal. At this time, the fixed codebook searching section 18 performs an error minimization operation in order to search for an optimal vector in the fixed codebook like the adaptive codebook searching section 17. Therefore, the fixed codebook searching section 18 receives LPCs before and after quantization from the LPC-LSP converting section 13 and LSP-LPC converting section 15, respectively.

**At page 7, last paragraph, starting on last line, please change to read as follows:**



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According to the present invention, it is paid attention to that in voice encoding, a reproduction characteristic does not degrade so much in the case of a vowel even if there is only a small number of encoding bits in a fixed codebook [is] and by lowering the encoding bit rate when the voice signal is a vowel, the average encoding bit rate can be lowered even when a voice part is sounded. Therefore, compared with the conventional case where the encoding bit rate is lowered only when a voiceless part is sounded, a bit rate needed for voice transmission can be further lowered while the quality of reproduced voice is maintained.

**At page 30, paragraph 1, starting on line 3, please change to read as follows:**

[AT] <u>At</u> the head of a voice part, the rate determining section judges that the voice signal is voice. In a subsequent frame, vowel spectrum components continue. In this case, since the power related to a fixed codebook is low, there is no influence in voice quality even if the number of bits of the fixed codebook is reduced. Therefore, rate information is modified from the full rate to half the rate.

<u>IN THE CLAIMS:</u>

Please amend the claims as follows:

2.(amended) [The device according to claim 1, further comprising:]

<u>A device for a variable-rate encoding system, comprising:</u>

<u>a judging unit judging whether a voice signal is a vowel when a voice part of a voice signal is sounded;</u>

<u>a rate setting unit setting a voice encoding bit rate to a bit rate lower than the bit rate usually used when the voice part is sounded if the voice signal is a vowel;</u>

an LSP coefficient calculating unit calculating an LSP coefficient obtained from the voice signal; and

an LSP interval judging unit judging whether an interval between the LSP coefficients is equal to or less than a prescribed threshold value.

6.(amended) [The method according to claim 5, further comprising:]

<u>A rate control method for a variable-rate encoding system, comprising</u>:

<u>(a) judging whether a voice signal is a vowel when a voice part of a voice signal is sounded;</u>

<u>(b) setting a voice encoding bit rate to a bit rate lower than the bit rate usually used when the voice part is sounded if the voice signal is a vowel;</u>

(c) calculating unit calculating an LSP coefficient obtained from the voice signal; and

(d) judging whether an interval between the LSP coefficients is equal to or less than a prescribed threshold value.

10.(amended) [The storage medium according to claim 9, the process further comprising:]

<u>A computer-readable storage medium which records a program for enabling a computer to implement a rate control method for a variable-rate encoding system, the process comprising:</u>

<u>(a) judging whether a voice signal is a vowel when a voice part of a voice signal is sounded;</u>

(b) setting a voice encoding bit rate to a bit rate lower than the bit rate usually used when the voice part is sounded if the voice signal is a vowel;

(c) calculating unit calculating an LSP coefficient obtained from the voice signal; and

(d) judging whether an interval between the LSP coefficients is equal to or less than a prescribed threshold value.